AMENDMENTS TO THE CLAIMS

Claim 1 (previously presented): A computerized, Internet protocol (IP) based voice response system for servicing a call received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network;

an IP network medium connected to the gateway;

a network server in communication with the IP network medium for automated interaction with a user participating in the call; and

a configuration server for performing a blasting process to provide automated dynamic management of the network server.

Claim 2 (original): The voice response system of claim 1, wherein the network server comprises a host computer for executing a voice application program, a grammar database corresponding to a set of recognizable utterances, and a voice recognition engine for comparing a speech input from the user against the set of recognizable utterances.

Claim 3 (original): The voice response system of claim 2, wherein the voice application program is a VoiceXML program.

Claim 4 (original): The voice response system of claim 2, further comprising a firewall in communication with the network medium for connecting the network server to an external IP network through the firewall, wherein the voice application program is remotely hosted on the external IP network.

Claim 5 (original): The voice response system of claim 2, wherein the network server performs call control communications with the PSTN-to-IP gateway in accordance with a SIP protocol.

Claim 6 (previously presented): A scalable, computerized, Internet protocol (IP) based voice response system for servicing a plurality of calls received over a public switched telephone network (PSTN) comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network;

an IP network medium connected to the gateway;

a plurality of network servers in communication with the network medium for automated interaction with a set of users participating in the plurality of calls; and

a proxy server in communication with the PSTN-to-IP gateway for load balancing the plurality of calls and providing differentiated and targeted service control for the plurality of calls amongst the plurality of network servers.

Claim 7 (original): The voice response system of claim 6, wherein each network server of the plurality of network servers comprises a host computer having a distinct network identification number.

Claim 8 (original): The voice response system of claim 7, further comprising a configuration server for automatically loading and configuring an initial software environment for the host computer during its initial bootup sequence based upon the network identification number.

Claim 9 (previously presented): A method of using voice over Internet protocols (VoIP) to handle circuit switched calls in a voice activated system, the method comprising:

terminating a circuit switched call at a conversion device that translates the circuit switched call into a VoIP format as a packet switched call;

applying differentiated and targeted service control to the packet switched call to forward the packet switched call in the VoIP format from the conversion device to a computer system; and

performing speech recognition on the call using audio data extracted from the VoIP format by the computer system.

Claim 10 (original): The method of claim 9, wherein the conversion device and the computer system are located in close physical proximity.

Claim 11 (original): The method of claim 9, wherein there is a second computer system physically distant from the conversion device and wherein the forwarding goes to the second computer system responsive to a failure of the first computer system.

Claim 12 (original): The method of claim 9, further comprising prior to the forwarding sending a message from the conversion device to a second computer system, the second computer system selecting the computer system from a plurality of computer systems to receive the call.

Claim 13 (original): The method of claim 12, wherein the selecting according to a predetermined set of criteria to balance number of calls being handled by each of the plurality of computer systems.

Claim 14 (original): The method of claim 12, wherein the message comprises a session initiation protocol (SIP) request.

Claim 15 (original): The method of claim 12, wherein the forwarding occurs responsive to a SIP acknowledgement from the computer system.

Claim 16 (previously presented): The voice response system of claim 1, further including a proxy server, wherein the configuration server directs the proxy server how to perform call discrimination.

Claim 17 (previously presented): The voice response system of claim 16, wherein call discrimination can allow calls on an entity basis.

Claim 18 (previously presented): The voice response system of claim 16, wherein call discrimination can disallow calls on an entity basis.

Claim 19 (previously presented): The voice response system of claim 16, wherein the configuration server provides re-purposing of at least one of the proxy server and the network server.

Claim 20 (previously presented): The voice response system of claim 1, further including a proxy server, wherein if the proxy server detects that a number of calls exceeds a predetermined threshold, then the proxy server follows at least one predetermined call routing rule provided by the configuration server.

Claim 21 (previously presented): The voice response system of claim 20, wherein the predetermined call routing rules include sending a busy signal to a first entity and allowing calls to a second entity.

Claim 22 (previously presented): A computerized, Internet protocol (IP) based voice response system for servicing a call received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network;

an IP network medium connected to the gateway;

a network server in communication with the IP network medium for automated interaction with a user participating in the call; and

a proxy server in communication with the IP network medium and the network server, wherein the proxy server provides differentiated and targeted service control for the call.

Claim 23 (previously presented): The voice response system of claim 22, wherein the differentiated and targeted service control can allow calls on a per client basis.

Claim 24 (previously presented): The voice response system of claim 22, wherein the differentiated and targeted service control can disallow calls on a per client basis.

Claim 25 (previously presented): The voice response system of claim 22, wherein if the proxy server detects that a number of calls exceeds a predetermined threshold, then the differentiated

and targeted service control includes following predetermined call routing rules.

Claim 26 (previously presented): The voice response system of claim 25, wherein the predetermined call routing rules include sending a busy signal to a first client and allowing calls to a second client.

Claim 27 (previously presented): The voice response system of claim 22, wherein the network server and proxy server are dynamically reconfigurable.

Claim 28 (previously presented): The voice response system of claim 27, wherein dynamic reconfiguration includes mapping a new software configuration.

Claim 29 (previously presented): A computerized, Internet protocol (IP) based voice response system for servicing a call received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network;

an IP network medium connected to the gateway;

a network server in communication with the IP network medium for automated interaction with a user participating in the call; and

means for providing differentiated and targeted service control over the call in operative relation to the IP network medium and the network server.

Claim 30 (previously presented): The voice response system of claim 29, wherein the differentiated and targeted service control can allow calls on a per client basis.

Claim 31 (currently amended): The voice response system of claim 29, wherein the differentiated and targeted service control can disallow calls on a per client basis.

Claim 32 (currently amended): The voice response system of claim 29, wherein if a number of calls exceeds a predetermined threshold, then the means for providing the differentiated and targeted service control follows predetermined call routing rules.

Claim 33 (previously presented): The voice response system of claim 32, wherein the predetermined call routing rules include sending a busy signal to a first client and allowing calls to a second client.

Claim 34 (previously presented): The voice response system of claim 29, wherein the means for providing the differentiated and targeted service control can be dynamically reconfigured.

Claim 35 (previously presented): The voice response system of claim 34, wherein dynamic reconfiguration includes a new setup of the proxy server.

Claim 36 (previously presented): A computerized, Internet protocol (IP) based voice response system for servicing a call received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network;

an IP network medium connected to the gateway;

a network server in communication with the IP network medium for automated interaction with a user participating in the call;

a configuration server for providing automated dynamic management of the network server; and

a proxy server, wherein if the proxy server detects that a number of calls exceeds a predetermined threshold, then the proxy server follows at least one predetermined call routing rule provided by the configuration server.

Claim 37 (previously presented): The voice response system of claim 36, wherein the predetermined call routing rules include sending a busy signal to a first entity and allowing calls to a second entity.

Claim 38 (previously presented): A computerized, Internet protocol (IP) based voice response system for servicing a call received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network;

an IP network medium connected to the gateway;

a network server in communication with the IP network medium for automated interaction with a user participating in the call; and

a proxy server in communication with the IP network medium and the network server, wherein the proxy server provides call discrimination,

wherein if the proxy server detects that a number of calls exceeds a predetermined threshold, then call discrimination includes following predetermined call routing rules.

Claim 39 (previously presented): The voice response system of claim 38, wherein the predetermined call routing rules include sending a busy signal to a first client and allowing calls to a second client.

Claim 40 (previously presented): A computerized, Internet protocol (IP) based voice response system for servicing a call received over a public switched telephone network (PSTN), the voice response system comprising:

a PSTN-to-IP gateway for connecting to the public switched telephone network;

an IP network medium connected to the gateway;

a network server in communication with the IP network medium for automated interaction with a user participating in the call; and

means for providing call discrimination over the call in operative relation to the IP network medium and the network server,

wherein if a number of calls exceeds a predetermined threshold, then the means for providing call discrimination follows predetermined call routing rules.

Claim 41 (previously presented): The voice response system of claim 40, wherein the predetermined call routing rules include sending a busy signal to a first client and allowing calls to a second client.

Claim 42 (new): A call handling system comprising:

a packet network;

a network interface for interfacing the packet network to at least one external network; and

an application server for sending signaling information to the network interface via the packet network, the signaling information indicating a call handling function for a call received by the network interface.

Claim 43 (new): The call handling system of claim 42, wherein the network interface further comprises means for transmitting the signaling information to the at least one of the external networks to execute the call handling function.

Claim 44 (new): The call handling system of claim 42, wherein the call handling function includes one of an outbound routing of the call, an outbound transfer of the call and a rejection of the call.

Claim 45 (new): The call handling system of claim 42, wherein the network interface further comprises means for selecting the application server from a plurality of application servers.

Claim 46 (new): The call handling system of claim 45, wherein the means for selecting the application server comprises a SIP proxy server.

Claim 47 (new): The call handling system of claim 45, wherein the means for selecting the application server comprises means for establishing a session with the application server and

establishing a media stream between the application server and the network interface.

Claim 48 (new): The call handling system of claim 47, wherein the media stream is controlled by the real time transport protocol ("RTP").

Claim 49 (new): The call handling system of claim 42, wherein the call handling function is an outbound transfer of the call and the network interface establishes a connection with a third party as a function of the signaling information.

Claim 50 (new): The call handling system of claim 42, wherein the packet network is a voice over Internet protocol ("VoIP") network.

Claim 51 (new): The call handling system of claim 42, wherein the signaling information is generated using the session initiation protocol ("SIP").

Claim 52 (new): The call handling system of claim 42, wherein the at least one external network is a TDM ("Time Division Multiplexing") network.

Claim 53 (new): The call handling system of claim 42, wherein the at least one external network is a second packet network.

Claim 54 (new): A method operating a call handling system, the method comprising:

receiving a call at a network interface from an external network;

selecting an application server to handle the call; and

sending a first set of signaling information from the application server to the network interface via a packet network, the first set of signaling information indicating a call handling function for the call.

Claim 55 (new): The method of claim 54, further comprising transmitting a second set of signaling information from the network interface to the external network in response to the first set of signaling information, the second set of signaling information causing the call handling function to be executed on the call.

Claim 56 (new): The method of claim 54, wherein selecting the application server comprises sending a second set of signaling information from the network interface to the application server via the packet network.

Claim 57 (new): The method of claim 54, wherein the call handling function includes one of an outbound routing of the call, an outbound transfer of the call and a rejection of the call.

Claim 58 (new): The method of claim 54, further comprising selecting the application server comprises selecting the application server from a plurality of application servers using a SIP proxy server.

Claim 59 (new): The method of claim 54, wherein selecting the application server comprises:

establishing a session with the application server; and

establishing a media stream between the application server and the network interface.

Claim 60 (new): The method of claim 59, wherein the media stream is controlled by the real time transport protocol ("RTP").

Claim 61 (new): The method of claim 54, wherein the call handling function is an outbound transfer of the call, and wherein the method further comprises establishing a connection between the network interface and a third party as a function of the first set of signaling information.

Claim 62 (new): The method of claim 54, wherein the packet network is a voice over Internet protocol ("VoIP") network.

Claim 63 (new): The method of claim 54, wherein sending the first set of signaling information comprises generating the first set of signaling information using the session initiation protocol ("SIP").

Claim 64 (new): The method of claim 54, wherein the external network is a TDM ("Time Division Multiplexing") network.

Claim 65 (new): The method of claim 54, wherein the external network is a packet network.

Claim 66 (new): A call handling system comprising:

a packet network;

a network interface for receiving a call from an external network and generating a first set of signaling information in response to the call;

an application server for receiving the first set of signaling information via the packet network, for establishing a session with the network interface in response to the first set of signaling information, and for sending a second set of signaling information to the network interface via the packet network, the second set of signaling information indicating a call handling function to be performed on the call.